Amendments To The Specification:

In the English translation document, please add the section heading and paragraph at page 1 written line 3, after the title, as follows:

--CROSS REFERENCE TO RELATED APPLICATIONS

This application is the US National Stage of International Application No. PCT/EP2004/050740, filed May 10, 2004 and claims the benefit thereof. The International Application claims the benefits of European application No. 03018497.2 EP filed August 14, 2003, both of the applications are incorporated by reference herein in their entirety.--

In the English translation document, please add the section heading and paragraph page 1 written line 3, after the newly added CROSS REFERENCE TO RELATED APPLICATIONS section, as follows:

--FIELD OF INVENTION

The present invention relates to call re-direction for an SIP telephone number of an SIP client in a combined wired and packet-switched network.--

In the English translation document, please add the section heading at page 1 written line 3, after the newly added FIELD OF INVENTION section, as follows:

--BACKGROUND OF INVENTION--

In the English translation document, please add the section heading at page 2 written line 20, as follows

--SUMMARY OF INVENTION--

In the English translation document, please amend the paragraphs at page 2 written lines 20-23, as follows:

An~~The~~ object of the present invention is to make possible and to set up a call re-direction between SIP and PSTN networks.

This object is achieved by the features of the method in accordance with ~~claim 1~~the independent claim.

In the English translation document, please add the section heading at page 4 written line 6, as follows:

--BRIEF DESCRIPTION OF THE DRAWINGS--

In the English translation document, please add the section heading at page 4 written line 19, as follows:

--DETAILED DESCRIPTION OF INVENTION--

In the English translation document, please amend the paragraph at page 4 written lines 19-29 through page 5 written lines 1-8, as follows:

Figure 1 shows an arrangement already described in the introduction of a combination of PSTN and SIP network. Figure 2 shows a Voice-over-IP network VoIP, with two Media Gateway Controllers ~~MCC~~<u>MGC</u> A, with the assigned domain mgca.munich.de, and MGC B, with the assigned domain mgcb.miesbach.de These two Media Gateway Controllers communicate with each other by means of an IP connection through the SIP T protocol. The Media Gateway Controller MGC A further controls by means of an IP connection and by the Media Gateway Control Protocol, abbreviated to MGCP, a Media Gateway MG A. This Media Gateway MG A is connected via a Time Division Multiplex connection, abbreviated to TDM, to first "classical" PSTN switching equipment PSTN/ISDN1. This PSTN/ISDN1 switching equipment in its turn has a connection via the Signaling System 7, abbreviated to SS7, or ISDN User Part protocol, abbreviated to ISUP protocol, to the Media Gateway Controller MGC A. Two PSTN telephones PSTN Phone A and PSTN Phone C are connected to the first switching equipment PSTN/ISDN1 for example.

In the English translation document, please amend the paragraph at page 6 written lines 19-30, as follows:

The second PSTN switching equipment PSTN/ISDN2 detects the specific numerical sequence or identifier, evaluates this and the SIP telephone number and then sends an ISUP message, such as ISUP:IAM, with a special registration code, the SIP telephone number and telephone number of the subscriber connection at the Media Gateway Controller MGC B. The Media Gateway

Controller MGC B evaluates this message and then sends an SIP:REGISTER message with the SIP telephone number, the SIP domain, the telephone number of the PSTN connection and its own SIP domain to the SIP Registrar SIP RA. ~~I~~i.e.

From: sip:+49199462518@sip.munich.de

Contact:<sip:+498024773377@mgcb.miesbach.de>

In the English translation document, please amend the paragraphs at page 7 written lines 13-31 through page 8 written lines 1-7, as follows:

If a second subscriber now wants to contact a first subscriber from an SIP client and dials the SIP telephone number of the first subscriber, an SIP:INVITE message is sent from the SIP client of the second subscriber to the SIP proxy SIP PA. Such as:

INVITE sip:+49199462518@sip.munich.de SIP/2.0

From: client02@sip.munich.de;tag=1c24841

To: sip:+49199462518@sip.munich.de

. . . .

The SIP proxy SIP PA now searches through the Location Service database LSA, to determine the current contact address or telephone number of the desired SIP telephone number. After determination of the current telephone number

~~498024773577~~498024773377@mgcb.miesbach.de

The SIP proxy SIP PA now searches through the Location Service database LSA, to determine the current contact address or telephone number of the desired SIP telephone number. After determination of the current telephone number

498024773577@mgcb.miesbach.de

the SIP Proxy SIP PA modifies the SIP:INVITE message by entering the new telephone number, to:

INVITE sip:+~~498024773377498024773577~~498024773377@mgcb.miesbach.de SIP/2.0

From: client02@sip.munich.de;tag=1c24841

To: sip:+49199462518@sip.munich.de

. . . .

and sends this to the Media Gateway Controller MGC B.

The Media Gateway Controller MGC B evaluates this message, detects the PSTN telephone number in the SIP:INVITE message and then sends an ISUP message to the second switching equipment PSTN/ISDN2. This evaluates the ISUP message and builds a call to the PSTN telephone PSTN Phone B.

In the English translation document, please amend the paragraphs at page 8 written lines 26-32 through page 9 written lines 1-28, as follows:

The third subscriber calls the SIP telephone number of the first subscriber from the PSTN telephone PSTN Phone C. The first switching equipment PSTN/ISDN1 then sends an ISUP message with the desired telephone number and the telephone number of the calling subscriber connection, that is of the PSTN telephone PSTN Phone C, to the Media Gateway Controller MGC A. The Media Gateway Controller MGC A evaluates this message and sends an SIP:INVITE message with the called and the calling telephone number to the SIP Proxy SIP PA. The domain of the desired SIP telephone number will be supplemented automatically in this case by the Media Gateway Controller. It can be permanently administered in the Media Gateway Controller or in the routing database of the Media Gateway Controller. Such as:

INVITE sip:+49199462518@sip.munich.de SIP/2.0

From: +~~49897224996~~498972224996@mgca.munich.de;tag=23d21

To: sip:+49199462518@sip.munich.de

. . . .

The SIP Proxy SIP PA evaluates this message and sends a request to the Location Service database LSA, in order to obtain the desired SIP telephone number or the current address or telephone number. After successfully determining the desired telephone number the SIP Proxy SIP PA modifies the SIP:INVITE message by entering the current telephone number of the desired SIP subscriber and sends it to the domain of the telephone number determined, that is to the Media Gateway Controller MGC B. For example:

INVITE sip:+498024773377@mgcb.miesbach.de SIP/2.0

From: +~~498972224996Omgca~~498972224996@mgca.munich.de;tag=23d21

To: sip:+49199462518@sip.munich.de

. . . .